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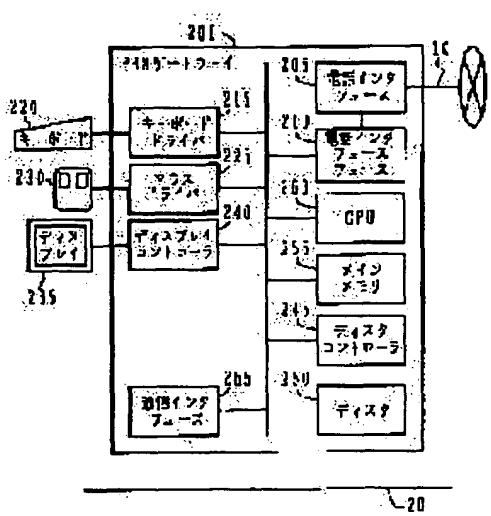
(54) AUDIO COMMUNICATION EQUIPMENT

(57)Abstract:

PROBLEM TO BE SOLVED: To provide the sounds of much better quality by varying the set value of the storage amount of audio data to be stored in a storage means during sound repetition.

SOLUTION: A telephone interface driver 210 has an internal buffer, stores audio data sent from a telephone line 10 through a telephone interface 205 and stores audio data sent through the telephone interface 205 to the telephone line 10. When the operator of a personal computer(PC) feels the delay of sounds is increased during the reception of a telephone call from a party, a sound repetition setting picture is displayed on that display screen and the set value of "'audio data storage amount' is reduced. Thus, the delay amount of sounds to be reproduced is reduced. When sounds are interrupted, the sound repetition setting picture is displayed on the display screen of the PC and the set value of 'audio data storage amount' is increased. Thus, sounds to be reproduced are prevented from being interrupted.



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* NOTICES *

CLAIMS

[Claim(s)]
[Claim 1] Speech-communication equipment characterized by having an accumulation means to accumulate temporarily after receiving the voice data sent from other speech-communication equipments, a reproduction means to reproduce the voice data sent from this accumulation means by the voice output means, and the set point adjustable means that carries out adjustable [of the set point of the accumulated dose of the voice data accumulated for the above-mentioned accumulation means] into voice relay.
[Claim 2] According to the Dial Tone Multi Frequency to which the above-mentioned set point adjustable means has been sent from telephone in speech-communication equipment according to claim 1, it is speech-communication equipment characterized by carrying out adjustable [of the set point of the accumulated dose of voice data].
[Claim 3] It is speech-communication equipment characterized by to have the control means which control the accumulated dose of the voice data accumulated for the above-mentioned accumulation means in speech-communication equipment according to claim 1, to be sent after being sent from other speech-communication equipments and accumulating voice data for the above-mentioned accumulation means, when the accumulated dose of these control means of the voice data accumulated at the above-mentioned accumulation means became less than the predetermined value, and to accumulate again a part of the same voice data as voice data for the above-mentioned accumulation means.

[Translation done.]

* NOTICES *

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2.**** shows the word which can not be translated.
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DETAILED DESCRIPTION

[Detailed Description of the Invention]
[0001]
[The technical field to which invention belongs] this invention exchanges voice data between the speech-communication equipment which is terminals, and relates to the speech-communication equipment used for the voice relay system which can do conversation.
[0002]
[Description of the Prior Art] As a voice relay system which telephones by transmitting and receiving voice data in the personal computers (personal computer) connected to LAN (Local Area Network), Talkware (registered trademark) of Hitachi indicated by the Nikkei data pro data COM, 1995.3, and p.681 is known, for example. Moreover, as a voice relay system which telephones using the Internet, they are Internetworking, 1996.9, and Netscape indicated by p.14-15, for example. Cool of Communications Talk (registered trademark) is known.
[0003] When transmitting and receiving voice data on a communication network, voice data is overdue with the load of a communication network, and a partner is reached. A sound piece will be generated, when voice data will arrive behind time, if it is made to reproduce immediately when voice data is received. Then, Talkware and Cool It is made to prevent a sound piece in Talk by storing slight voice data which received and reproducing.
[0004] Especially, in Talkware, the amount which stores voice data is calculated automatically, and voice is regenerated.
[0005]
[Problem(s) to be Solved by the Invention] However, if the transmission speed of a network becomes slow, since the accumulated dose of voice data will also increase, audio delay becomes large. If audio delay becomes large, some operators who have received may memorize sense of incongruity to the voice which has received. Moreover, a sound piece will be generated, when the transmission lag of voice data which exceeds this upper limit occurs, since a upper limit is prepared in the accumulated dose of voice data and it is in it. Since the voice which has received becomes intermittent with PUTSUPUTSU, a sound piece may memorize displeasure to the voice which has received for some operators who have received.
[0006] When memorizing displeasure in the case where sense of incongruity is memorized to audio delay, or a sound piece, there are individual differences by the operator. Therefore, the amount which stores voice data was automatically calculated like before, and there was a problem that it was difficult to depend according to an operator's sensibility and to offer the voice of good tone quality, by the method of regenerating voice. Since delay may become large since a sound piece may be somewhat generated for some operators when saying that it is made to want to want to lessen audio delay or, it may be said that he wants to lose a sound piece.
[0007] The purpose of this invention is to offer the speech-communication equipment which can offer the voice of better tone quality.
[0008]
[Means for Solving the Problem] (1) this invention is equipped with an accumulation means to accumulate temporarily after receiving the voice data sent from other speech-communication equipments, a reproduction means to reproduce the voice data sent from this accumulation means by the voice output means, and the set point adjustable means that carries out adjustable [of the set point of the accumulated dose of the voice data accumulated for the above-mentioned accumulation means] into voice relay in order to attain the above-mentioned purpose. By this composition, since it can carry out adjustable [of the set point of the accumulated dose of voice data] into voice relay by judgment of a user, the state which is easy to hear it for a user, i.e., the voice of better tone quality, can be offered.
[0009] (2) In the above (1), the above-mentioned set point adjustable means is made to carry out adjustable [of the set point of the accumulated dose of voice data] preferably according to the Dial Tone Multi Frequency sent from

telephone. By this composition, even from telephone, since it can carry out adjustable [of the set point of the accumulated dose of voice data] into voice relay, the state of being easy to hear it for a user can be offered.

[0010] (3) It has the control means which control preferably the accumulated dose of the voice data accumulated for the above-mentioned accumulation means in the above (1), and when the accumulated dose of the voice data accumulated at the above-mentioned accumulation means became less than the predetermined value, after being sent from other speech-communication equipments and accumulating voice data for the above-mentioned accumulation means, these control means are sent and accumulate again a part of same voice data as voice data for the above-mentioned accumulation means. By this composition, while being able to prevent a sound piece, the tone quality of the sound reproduced for sound piece prevention may be improved.

[0011]

[Embodiments of the Invention] Hereafter, the voice relay system using the speech-communication equipment by 1 operation gestalt of this invention is explained using drawing 1 - drawing 13 . First, drawing 1 is used and the whole voice relay-system composition by 1 operation gestalt of this invention is explained.

[0012] It connects with the telephone line 10 and the PBX gateway 200 can receive the telephone got from the telephone line 10. It connects with the personal computer (personal computer) 300,400 which constitutes the speech-communication equipment by this operation form through LAN (Local AreaNetwork)20, and the information about the PBX gateway 200 can be exchanged by LAN20 course. The PBX gateway 200 can send audible-sound voice to a personal computer 300,400 from a telephone, or can reproduce the voice data sent from the personal computer 300,400 to the telephone line.

[0013] The hand set 390,490 is connected to the personal computer 300,400, and audio I/O is possible using a hand set 390,490. In the state where can display setting screen 300A on the display of a personal computer 300, and this setting screen 300A is displayed on it, change of the set point of a personal computer 300 is possible. About the contents of this setting screen 300A, and the method of setting change, it mentions later using drawing 12 .

[0014] Next, the hardware composition of the PBX gateway in the voice relay system by 1 operation form of this invention is explained using drawing 2 .

[0015] It connects with the telephone line 10 and the telephone interface 205 in the PBX gateway 200 can recognize telephone arrival etc. The telephone interface 205 is controlled by the telephone interface driver 210. The telephone interface driver 210 accumulates the voice data which has the internal buffer which is not illustrated, and accumulates the voice data sent through the telephone interface 205 from the telephone line 10, and is sent out to the telephone line 10 through the telephone interface 205.

[0016] The keyboard driver 215 receives the input from a keyboard 220. The mouse driver 225 receives the input from a mouse 230. The display to a display 235 is controlled by the display controller 240. The program performed by the PBX gateway 200 is read into main memory 255 from a disk 250 by the disk controller 245, and is performed by CPU260. The PBX gateway 200 is connected by the communication interface 265 in LAN20 which is a communication network.

[0017] Next, the hardware composition of the personal computer which is speech-communication equipment in the voice relay system by 1 operation form of this invention is explained using drawing 3 . In addition, although explained taking the case of a personal computer 300, the personal computer 400 also has same composition here.

[0018] It connects with the hand set 390 and the voice-input/output interface 305 can carry out the voice input/output of a hand set 390. The voice-input/output interface 310 is controlled by the voice-input/output interface driver 310. The voice interface driver 310 accumulates the voice data which has the internal buffer which is not illustrated, and accumulates the voice data sent through the voice interface 305 from a hand set 390, and is sent out to a hand set 390 through the voice interface 305.

[0019] The keyboard driver 315 receives the input from a keyboard 320. The mouse driver 325 receives the input from a mouse 330. The display to a display 335 is controlled by the display controller 340. The program performed with a personal computer 300 is read into main memory 355 from a disk 350 by the disk controller 345, and is performed by CPU360. The personal computer 300 is connected by the communication interface 365 in LAN20 which is a communication network.

[0020] Next, processing of the voice relay of the PBX gateway 200 in the voice relay system by 1 operation form of this invention is explained using drawing 4 and drawing 5 .

[0021] The voice relay program performed by the PBX gateway 200 is read into main memory 255 from a disk 250 by the disk controller 245, and is performed by CPU260. A voice relay program is installed in a disk 250 from storages, such as FD and CD-ROM.

[0022] In Step 410 of drawing 4 , CPU260 of the PBX gateway 200 will usually be in the state of the waiting for telephone arrival. A voice relay program investigates periodically whether the telephone interface 205 has telephone

arrival using the telephone interface driver 210 shown in drawing 2 at the time of an arrival-of-the-mail waiting state.
[0023] If there is telephone arrival, in Step 420, CPU260 will perform an audio response by opening the voice file saved on the disk 250, and sending the voice data currently recorded on the file to the voice interface 205 using the telephone interface driver 210. the contents of this audio response -- for example, "-- this is the PBX gateway Please push an extension number. They are the contents which announce making an extension number push like ".
[0024] And in Step 430, CPU260 receives an extension number. In the case of registration of an extension number, a voice relay program investigates periodically the number inputted into the telephone interface 205 by the Dial Tone Multi Frequency using the telephone interface driver 210 shown in drawing 2 , and acquires the Dial Tone Multi Frequency which the telephone interface 205 has recognized from the telephone interface driver 210.
[0025] If the recognized extension number is received, in Step 440, using the extension number-IP address correspondence table 500 shown in drawing 5 , CPU260 will acquire an IP address, will call it to the personal computer 300 connected by LAN20, and will send a message.
[0026] Here, the composition of the extension number-IP address correspondence table 500 is explained using drawing 5 . As shown in drawing 5 , the extension number-IP address correspondence table 500 consists of an extension number 510, a name 520, and IP address 530. For example, it is "taro-yamamoto" and IP address 530 of the personal computer currently used is "192.10.1.12" as the operator of the personal computer corresponding to "1234" in an extension number 510 is indicated by the column of a name 520. That is, when the extension number received in Step 430 is "1234", a corresponding IP address "192.10.1.12" is acquired. In addition, the extension number-IP address correspondence table 500 is stored in main memory 255. Moreover, with the personal computer which received the call from the PBX gateway 200 in Step 440, arrival of the mail is told to the user of a personal computer by displaying a call bell or a message. And a push of the button which receives a telephone of the user of this personal computer sends the message of Reception O.K. to the PBX gateway 200 from a personal computer.
[0027] In Step 450, if CPU260 judges whether a personal computer is Reception O.K. and the PBX gateway receives the message of Reception O.K., in Step 460, CPU260 will start voice relay. Moreover, in not being Reception O.K., in Step 490, CPU260 hangs up a telephone.
[0028] In Step 460, CPU260 is later mentioned about the detail of processing of voice relay using drawing 9 and drawing 10 , although voice relay is started.
[0029] The voice relay program of CPU260 is investigating telephone cutting periodically after the voice relay start using the telephone interface driver 210. a ****** [that CPU260 detected telephone cutting in Step 470] -- or it judges whether the cutting message from a personal computer 300 was received When telephone cutting is detected or the cutting message from a personal computer 300 is received, in Step 480, CPU260 ends voice relay. And in Step 490, CPU260 disconnects a telephone and returns to the telephone arrival waiting state in Step 410.
[0030] Next, processing of the voice relay when calling a personal computer 400 from the personal computer 300 which is speech-communication equipment in the voice relay system by 1 operation gestalt of this invention is explained using drawing 6 - drawing 8 .
[0031] The voice relay program performed with a personal computer 300 is read into main memory 355 from a disk 350 by the disk controller 345 shown in drawing 3 , and is performed by CPU360. A voice relay program is installed in a disk 350 from storages, such as FD and CD-ROM.
[0032] When starting voice relay, after a user inputs into dispatch screen 300B which showed the "IP address" to drawing 7 and specifies the partner who talks over the telephone, he pushes "dispatch" button. In Step 610, CPU360 checks that the dispatch button has been pushed. Next, in Step 620, CPU360 sends a dispatch message to other personal computers 400.
[0033] Next, in Step 630, CPU360 is called to the personal computer 400 connected by LAN20, and sends a message. Moreover, with the personal computer 400 which received the call from the personal computer 300 in Step 630, arrival of the mail is told to the user of a personal computer by displaying a call bell or a message. And a push of the button which receives a telephone of the user of this personal computer sends the message of Reception O.K. to a personal computer 300 from a personal computer 400.
[0034] In Step 640, if CPU360 judges whether a personal computer is Reception O.K. and a personal computer 300 receives the message of Reception O.K., in Step 650, CPU360 will start voice relay. Moreover, in not being Reception O.K., it returns to the telephone arrival waiting state in Step 610.
[0035] In Step 650, although CPU360 starts voice relay, if voice relay is started, in the personal computer 300,400 in voice relay, screen 300C in the voice relay shown in drawing 8 will be displayed.
[0036] a ****** [that, as for CPU360, the cutting button of screen 300C in voice relay was pushed in Step 660] -- or it judges whether the cutting message from a personal computer 340 was received When the depression of a cutting button is detected or the cutting message from a personal computer 400 is received, in Step 670, CPU360 ends voice

relay. And CPU360 disconnects a telephone and returns to the telephone arrival waiting state in Step 610.
[0037] Next, the processing at the time of the data transmission in the voice relay processing in the voice relay system by 1 operation gestalt of this invention and data reception is explained using drawing 9 - drawing 11 .
[0038] If voice relay is started in Step 460 of drawing 4 , or Step 650 of drawing 6 , the PBX gateway 200 or a personal computer 300 will start data transmitting processing in which voice data is transmitted to the partner (a personal computer 300 or personal computer 400) who shows drawing 9 , and the data reception when receiving the voice data sent by the partner who shows drawing 10 . In addition, the following explanation explains the case where data are transmitted to a personal computer 400 from a personal computer 300.
[0039] First, drawing 9 is used and explained about data transmitting processing. In Step 910, CPU360 checks the amount of the voice data which transmits using the voice-input/output interface driver 310. Here, the set point of voice relay is explained using drawing 11 . As shown in drawing 11 , it consists of a content of a setting, and the set point, and these contents of a setting and set points are stored in main memory 355. In the example shown in drawing 11 , "the length of voice data" is set as "120" bytes (Byte). Gather voice data for every predetermined byte count, it is made to transmit, and "the length of voice data" shows the length of the voice data which transmits at once here. 120 bytes is equivalent to the sound signal of the length of about 30 mS(s) in the length of voice data. Moreover, the "voice data accumulated dose" is set as "400-500" byte. Here, the "voice data accumulated dose" shows the amount which accumulates voice data, in order to prevent a sound piece. This detail is explained in Steps 1030 and 1040 of drawing 10 . The "coding method" is set as "ADPCM." "Volume" is set as "10" (arbitrary unit).
[0040] In Step 920, if the amount of voice data checked in Step 910 judges whether it is larger than the set point "120" byte (the length of the voice data sent to a partner at once) of the "length of voice data" shown in drawing 11 and does not fill the set point with CPU360, it returns to Step 910 and repeats the check of the amount of voice data.
[0041] If the voice data which should transmit exceeds the set point of "the length of voice data", in Step 930, CPU360 will take out voice data from the voice-input/output interface 310 using the voice-input/output interface driver 310.
[0042] And in Step 940, CPU360 sends the taken-out voice data to a partner's personal computer.
[0043] In Step 950, CPU360 judges whether it is the end of voice relay, and if it is not an end, it will repeat Step 940 from Step 910.
[0044] Next, the processing at the time of data reception is explained using drawing 10 . In addition, in a personal computer 400, although data reception is performed, since a personal computer 300 and a personal computer 400 are the same composition, they are explained here based on the composition of the personal computer 300 shown in drawing 3 .
[0045] In Step 1010, a partner's personal computer receives the voice data to which CPU360 was sent in Step 740 of drawing 9 .
[0046] Next, in Step 1020, CPU360 checks the accumulated dose of the voice data in the voice-input/output interface 310. In the composition of the personal computer 300 shown in drawing 3 , the voice data received through LAN20 and the communication interface 365 is once incorporated by the internal buffer in CPU360. Then, voice data is inputted into the internal buffer of the voice-input/output interface driver 310 within the internal buffer of CPU360. Although the capacity of the internal buffer of the voice-input/output interface driver 310 is 4KB, about 400-500 bytes of voice data is always accumulated at the internal buffer. Every fixed time and every 30mS, the voice-input/output interface driver 310 will send out voice data to the voice-input/output interface 305, and will reproduce sound from a hand set 390. If the voice data accumulated at the internal buffer of the voice-input/output interface driver 310 is lost, sound is no longer reproduced from a hand set 390, and it will be in the state of a sound piece. Then, in Step 1020, CPU360 checks the accumulated dose of the voice data in the voice-input/output interface driver 310.
[0047] Here, as set as the "voice data accumulated dose" shown in drawing 11 , if "400-500" KB, in Steps 1030 and 1040, the accumulated dose of the voice data in the voice-input/output interface 310 will judge whether to be [more] or it is fewer than the maximum of this set-up voice data accumulated dose than the minimum value.
[0048] In Step 1030, the accumulated dose of the voice data which has CPU360 in the voice-input/output interface driver 310 is judged to be fewer than the maximum of this set-up voice data accumulated dose, and sets to Step 1040. If CPU360 is judged that there are more accumulated doses of the voice data in the voice-input/output interface driver 310 than the minimum value of this set-up voice data accumulated dose, it will set to Step 1060. The draft of voice data is inputted into the internal buffer of the voice-input/output interface driver 310 from the internal buffer of CPU360. That is, when processing of Step 1060 is performed, the accumulated dose of the voice data in the internal buffer of the voice-input/output interface driver 310 is within the limits of [proper] 400-500 bytes. Then, as shown in drawing 11 , since "the length of voice data" is set up with "120" bytes, it inputs the voice data for 120 bytes into the internal buffer of the voice-input/output interface driver 310 from the internal buffer of CPU360 by the draft of voice data. On the other hand, since the voice-input/output interface driver 310 has sent out 120 bytes of voice data to the voice-

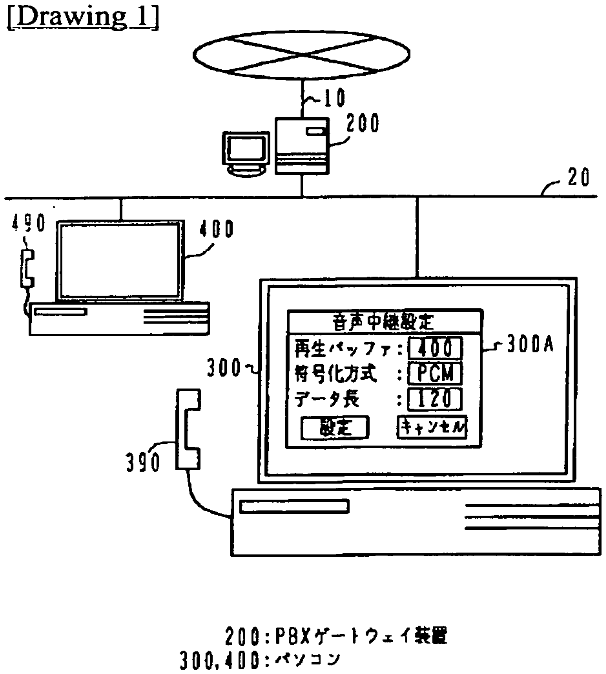
input/output interface 305 the fixed period, the accumulated dose of the voice data in the internal buffer of the voice-input/output interface driver 310 can maintain 400-500 bytes of proper range. Therefore, it is lost that the sound reproduced from a hand set 390 produces a sound piece.
[0049] In Step 1030, next, the accumulated dose of the voice data which has CPU360 in the voice-input/output interface driver 310 It is judged that it is fewer than the maximum of this set-up voice data accumulated dose, and it sets to Step 1040. If CPU360 is judged that there are few accumulated doses of the voice data in the voice-input/output interface driver 310 than the minimum value of this set-up voice data accumulated dose, it will set to Step 1050. 1.5 batches of voice data are inputted into the internal buffer of the voice-input/output interface driver 310 from the internal buffer of CPU360. That is, when processing of Step 1050 is performed, the accumulated dose of the voice data in the internal buffer of the voice-input/output interface driver 310 is the case where it becomes less than 400 bytes. Then, as shown in drawing 11 , since "the length of voice data" is set up with "120" bytes, it inputs the voice data for 180 bytes into the internal buffer of the voice-input/output interface driver 310 from the internal buffer of CPU360 by 1.5 batches of voice data. On the other hand, since the voice-input/output interface driver 310 has sent out 120 bytes of voice data to the voice-input/output interface 305 the fixed period, the accumulated dose of the voice data in the internal buffer of the voice-input/output interface driver 310 will increase one by one, and can return to 400-500 bytes which is a proper range. In addition, by 1.5 batches of voice data, in order to input the voice data for 180 bytes into the internal buffer of the voice-input/output interface driver 310 from the internal buffer of CPU360, 60 bytes of voice data of the half of 120 bytes of data of a draft and 120 bytes of this same data is sent. That is, 60 bytes of voice data will send the completely same data. Consequently, the sound reproduced from a hand set 390 with the voice-input/output interface 305 does not give displeasure for those for whom this uses a hand set 390, although the same sound will be repeated by 60 bytes. As the 1st reason, 60 bytes of voice data is about 15 mS(s) and very short time. Moreover, since the sound reproduced at this time is the same as front sound, unnaturalness does not occur on human being's acoustic sense. Here, although inputting false voice data into the voice-input/output interface 305 is also considered, since false voice data differs from the sound reproduced in front of it, on human being's acoustic sense, sense of incongruity will be sensed rather.
[0050] Therefore, since the accumulated dose of the voice data in the internal buffer of the voice-input/output interface driver 310 will increase one by one and can return to 400-500 bytes which is a proper range, it is lost. [of the sound reproduced from a hand set 390 producing a sound piece]
[0051] Moreover, in Step 1030, if CPU360 is judged that there are more accumulated doses of the voice data in the voice-input/output interface driver 310 than the maximum of this set-up voice data accumulated dose, it will return to the voice data reception in Step 1010, without carrying out anything. That is, if the accumulated dose of the voice data which is in the voice-input/output interface driver 310 in this case sends 500 bytes or more of voice data to the voice-input/output interface driver 310 here for a certain reason, voice data will increase and time will become this thing to reproduction of voice data. That is, delay of voice data becomes large and a partner's voice can be heard behind time.
[0052] Even when voice data arrives behind time by carrying out the accumulated dose of voice data into a certain set point, voice can be reproduced without way piece ****** and can lose [in / regeneration of voice data / as mentioned above] delay beyond preset value.
[0053] In addition, what is necessary is to transpose the voice-input/output interface 305 of the explanation mentioned above to the telephone interface 205 in the voice regeneration in the PBX gateway 200, although explanation of drawing 9 mentioned above and drawing 10 explained the voice regeneration in a personal computer, and just to transpose the voice-input/output interface driver 310 to the telephone interface driver 210, and the same voice regeneration can be performed also in the PBX gateway 200.
[0054] Next, processing of a voice relay setup is explained using drawing 1 and drawing 12 .
[0055] A push on the "setting" button of voice relay setting screen 300A shown in drawing 1 performs processing of a voice relay setup shown in drawing 12 .
[0056] In Step 1210, if it recognizes that the "setting" button of setting screen 300A was pushed, CPU360 will investigate which setup was changed, and will perform each processing.
[0057] In Step 1220, when CPU360 judges whether the value of a reproduction buffer is changed and the value of a reproduction buffer is changed, in Step 1230, CPU360 changes the "voice data accumulated dose" in the set point of voice relay shown in drawing 11 into the set-up value. In addition, when setting width of face is in a "voice data accumulated dose", setting width of face is made into "120" bytes in the example which showed the setting width of face to drawing 11 as what is decided beforehand. In voice relay setting screen 300A, since the lower limit of the set point of a "voice data accumulated dose" is set up, a "voice data accumulated dose" becomes 400-500 bytes. In this operation gestalt, the set point of a "voice data accumulated dose" can only be changed, and the amount of voice data accumulated at the voice-input/output interface driver 310 can be changed with the method explained by drawing 10 .

[0058] When it senses that delay of sound became large while receiving the telephone from a partner, as shown in drawing 1 , the operator of a personal computer 300 displays voice relay setting screen 300A on the display screen of a personal computer 300, and makes small the set point of a "voice data accumulated dose." By this, since a "voice data accumulated dose" can be made small, the amount of delay of the sound reproduced can be made small. Moreover, when a sound piece is generated, as shown in drawing 1 , voice relay setting screen 300A is displayed on the display screen of a personal computer 300, and the set point of a "voice data accumulated dose" is enlarged. By this, since a "voice data accumulated dose" can be enlarged, the sound piece of the sound reproduced can be prevented.
[0059] When memorizing displeasure in the case where sense of incongruity is memorized to audio delay, or a sound piece The case where it is said that it is made to want to want to lessen audio delay since there are individual differences and a sound piece may be somewhat generated for some operators by the operator, Or since delay may become large and it may say that he wants to lose a sound piece, it can depend according to an operator's sensibility and the voice of good tone quality can be obtained.
[0060] Next, in Step 1240, when it judges whether CPU360 had change in the coding method and a coding method has change, in Step 1250, CPU360 changes a coding method. Here, a coding method is a method which changes the voice inputted from the hand set 390 into the form which can be transmitted and received by LAN20. A change of a coding method is made by changing a setup of the voice-input/output interface 310 using the voice-input/output interface driver 310.
[0061] Moreover, in Step 1260, in CPU360, when it judges whether voice data length is changed and voice data length is changed, in Step 1270, CPU360 changes "the length of voice data." Only by changing this set point, the length of the voice data which transmits can be changed by processing at the time of the voice data transmission explained by drawing 9 .
[0062] As explained above, when a user can change a setup of voice relay, since a sound piece may be carried out, it can fulfill a request of those who want to carry out delay if possible, and a person with disagreeable carrying out a sound piece, since it may be delayed with this operation gestalt.
[0063] Next, the case where the set point of voice relay is changed is explained using terminals, such as telephone, using drawing 13 .
[0064] Since the personal computer is used as speech-communication equipment, although the set point can be changed in the example shown in drawing 12 using a display, in the case of speech-communication equipments, such as telephone without a display, the set point is used for the button for a setup etc., and it can set it up using the DTME (Dual Tone Multi Frequency) signal generated when the button of telephone is pushed.
[0065] Drawing 13 is flow chart **** explaining the processing whose PBX gateway 200 detects the Dial Tone Multi Frequency emitted from the telephone connected to the telephone network 10, and changes a setup.
[0066] In Step 1310, CPU260 of the PBX gateway 200 repeats and asks the telephone interface driver 210 a Dial Tone Multi Frequency into voice relay.
[0067] And in Step 1320, if it judges whether CPU260 detected the Dial Tone Multi Frequency and a Dial Tone Multi Frequency is detected, in Step 1330, CPU260 will acquire the Dial Tone Multi Frequency which received from the telephone interface driver 210.
[0068] Next, the acquired Dial Tone Multi Frequency judges whether it is an instruction of setting change. Here, the setting variation order is decided as a "voice data accumulated dose" will be set to "400", if "400" is pushed after being decided beforehand, for example, pushing "#."
[0069] And when a Dial Tone Multi Frequency is an instruction of setting change, in Step 1350, CPU260 changes the set point of voice relay shown in drawing 11 .
[0070] It comes to be able to perform a setup of the voice data accumulated dose of voice relay also from a personal computer to a telephone side with this operation form as mentioned above.
[0071]
[Effect of the Invention] According to this invention, the state which is easy to hear it for a user, i.e., the voice of better tone quality, can be offered now by judgment of a user.

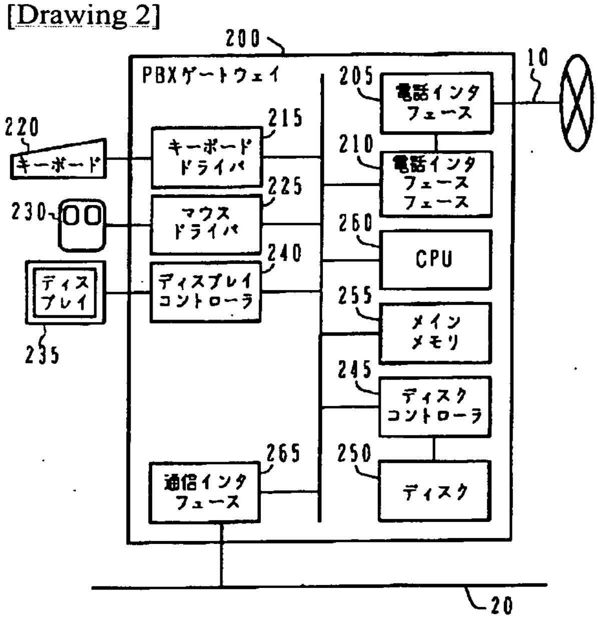
[Translation done.]

DRAWINGS

[Drawing 1]



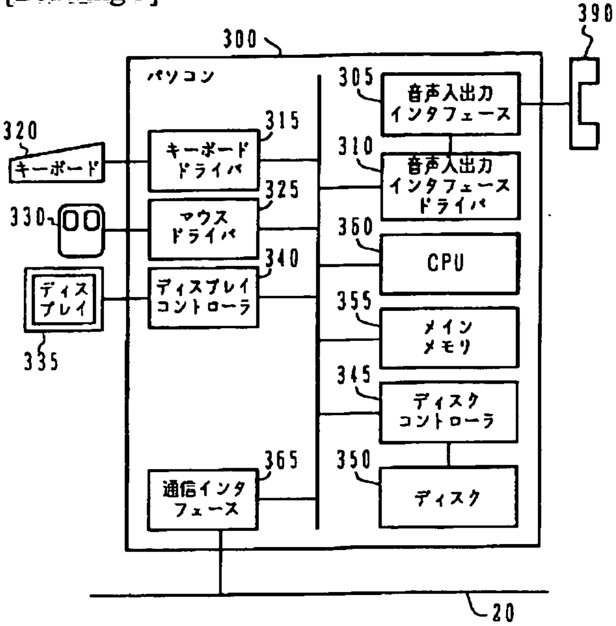
[Drawing 2]



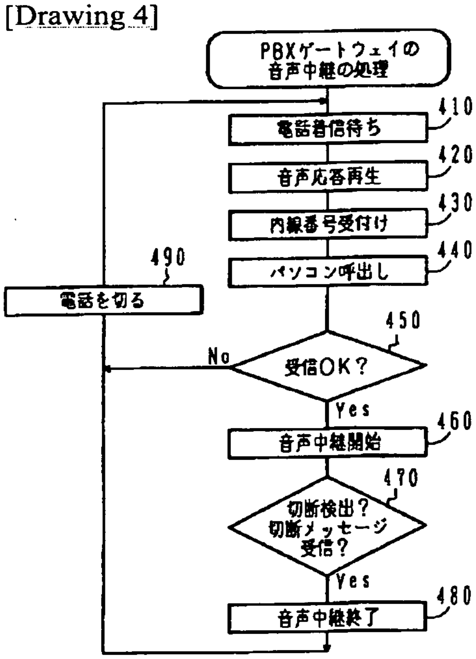
[Drawing 11]

設定内容	設定内容
音声データの長さ	100
音声データ蓄積量	400~500
符号化方式	PCM
音量	10

[Drawing 3]



[Drawing 4]



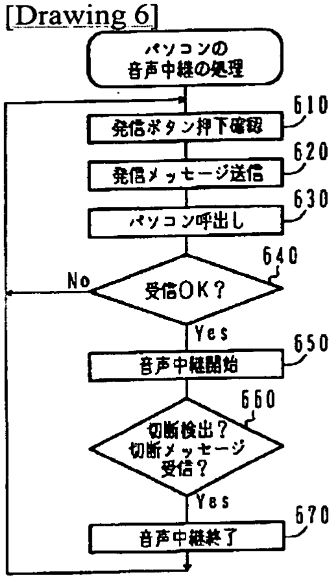
[Drawing 5]

500

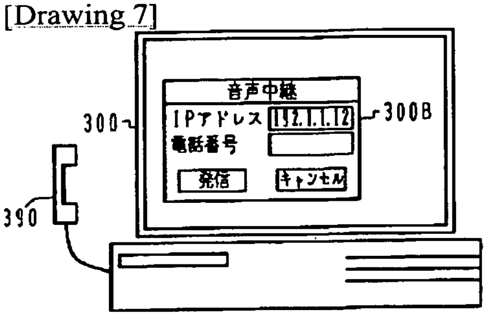
510 520 530

内線番号	氏名	IPアドレス
1234	taro-yamamoto	192.10.1.12
4567	jiro-tanaka	192.10.1.14
5678	saburo-sozuki	192.10.1.27
8976	ichiro-satoh	192.10.1.56
0000	root	192.10.1.1

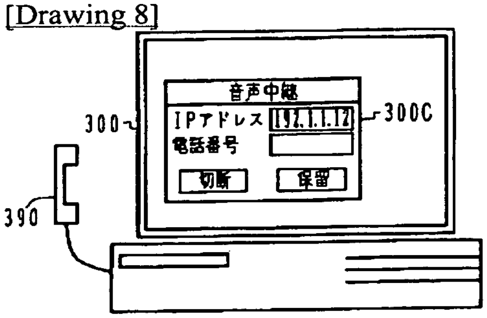
[Drawing 6]



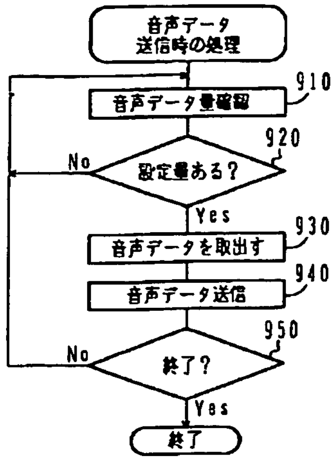
[Drawing 7]



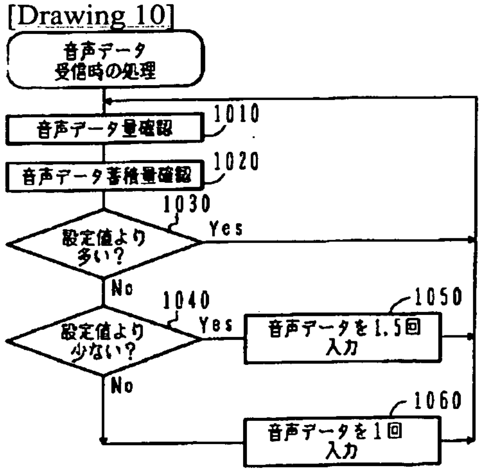
[Drawing 8]



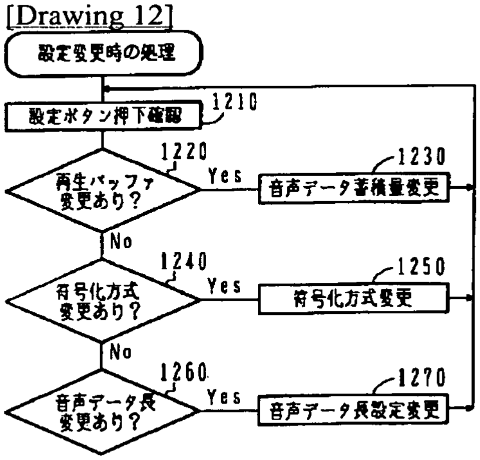
[Drawing 9]



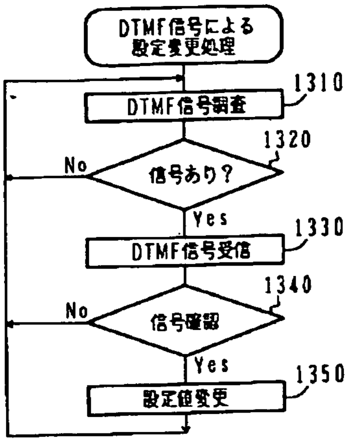
[Drawing 10]



[Drawing 12]



[Drawing 13]



[Translation done.]